REMARKS

Claims 1-27, 29-32 and 33-44 are pending in the present application. Applicant affirms election of the Group I, claims 1-27 and 29-32 drawn to a multi-channel interactive audio system. Claim 28 has been withdrawn and claim 26 has been cancelled. Claims 1,3,4,6,11,17,21,27,29 and 30 have been amended and new claims 33-44 have been added. Reconsideration of the claims is respectfully requested.

Claims 1-4,6,10,11,16-17 and 29-31 were rejected under 35 U.S.C. § 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3. Everett describes techniques for applying attenuation to audio signal frames in MPEG and similar compression systems. More specifically Everett teaches an on-screen display having features keyed to various audio sources. Six input tracks are stored as separate MPEG audio streams, with the user being enabled to mix these tracks in real time using volume control sliders 101-106 displayed on the workstation screen and moved by selection and dragging thereof with a cursor moved by a mouse or similar user-operable input device. The current setting of each of the sliders 101-106 determines the value of the LEVEL input to the respective attenuation stage 20 shown in FIG. 2 (Col. 5, ln 46-60). Further sliders control delay length in a reverberation loop and control the frequency spectrum and filter. (Col. 6, ln 6-11).

In reference to independent claims 1, 17, 21, 27, 29, 35 and 40, absent some specific teaching or suggestion in either Everett or ISO/IEC 11173-3 it would not have been obvious to one of ordinary skill in the art to store a plurality of compress audio components, partially unpack

and decompress the input frame, modify the scale factors in accordance with a user input, remix the subband data with the new scale factors, and recompress and repack them into an output frame. Everett teaches a computationally efficient method of mixing compressed audio signals without first completely decompressing the signals back into the time domain. Everett's mixed subband data can then be fed to a reconstruction filter to output the mixed audio signal. ISO/IEC 11173-3 teaches the compression of audio first presented in the PCM time domain, it does not teach how the input analysis can be bypassed to accept data already in partially encoded form, i.e. Everett's mixed subband data.

Furthermore, claims 1, 27 and 29 have been amended to specify "audio components having positional coordinates" (p. 7, ln 1-3, 14-16, 28), an "HID for receiving a positional input for a user" (p. 6, ln 26-27, p. 7, ln 3-4,8), and calculating the scale factors "in accordance with the positional input relative to the positional coordinates" (p. 9, ln 19-23). New claim 35 has similar limitations. Everett does not teach nor suggest calculation of the scale factors and thus mixing of the audio components in accordance with the relative *position* of the user to the audio components.

The rejection of independent claims 1, 17, 21, 27 and 29 and all their dependent claims is respectfully traversed.

In reference to claims 3 and 24, Applicant uses the scale factors to calculate the intra-subband masking effects, which includes both a relative comparison similar to Everett and a comparison to an absolute sound level. Everett merely discards data if indices (scale factors)

differ by more than a pre-programmed value. In reference to claims 4, 25, 30, Applicant "unpacks and decompresses *only* the subband data in the audible subbands". Everett extracts the MPEG streams group by group from the buffer and sub-band by sub-band. If the indices differ by more than a certain pre-programmed value, the group with the smaller index (larger scale-factor) is output and the other group is discarded. Everett reads out all of the subband data and then decides what data to discard. This is different and less efficient then the claimed approach.

In reference to claim 5, Everett (col. 3, ln 61-65) teaches "divided by scale factors" where as step c is specifically directed to "multiplied by the reciprocal of the maximum scale factor", which is a mathematically less intensive, hence faster operation. Davis (col.13, ln 49-50) describe a mantissa and an exponent factor that determine the number of bits by which to shift the subband data. This only permits division or multiplication by discrete powers of two. In claim 5, the subband scale factors determine the arbitrary scaling that is required, which is not limited to powers of two. Furthermore, the N-bit subband data is stored in a left shifted format in which the sign bit is aligned to the sign bit of the M-bit format. This feature is not taught nor suggested by the cited art.

Claim 6 had been amended to read "the input data frame further includes a header and a bit allocation table that are fixed from frame-to-frame so that only the content of the scale factors and subband data are allowed to vary but are otherwise of fixed size in the compressed stream." (p. 4, ln 25-27). Everett specifies "The final section SBD

carries the sub-band data as variable length fixed point values…" (Col 4, ln 16-17).

In reference to claim 7, Everett not only fails to disclose coding the subband data with fixed length codes but specifically teaches that "The final section SBD carries the sub-band data as variable length fixed point values…" (Col. 4, ln 16-17). Modification of Everett to use fixed length codes would render the system incompatible with MPEG audio streams, which use variable length codes to improve compression.

In reference to claim 8, Applicant uses the FLCs to extract the N-bit subband data, stores it as M-bit words with the leftmost bit a sign bit and left shifts the subband data until its sign bit is aligned with the M-bit sign bit leaving the M-N bits as noise. This is done to streamline the process of packing the compressed subband data, not to reallocate bits for noise considerations. Again Everett explicitly teaches the use of variable length codes (col. 3, ln. 40-42, col. 4, ln., 16-17). Furthermore, col. 4, ln 47-51 of Everett simply state that the streams are buffered but do not teach storing N-bit subband data in an M-bit word where the leftmost bit is a sign bit. Furthermore, ISO/IEC 11172-3 does not teach left shifting the data. Specifically, p. 74, ln 22-29 refers to the assignment of bits to subbands so that the effects of quantization noise are minimized.

In reference to claim 9, ROM 36 does not store the header and bit allocation table. ROM 36 is a pre-computed mixing LUT (look up table) that receives as inputs the difference in scale factors and subband data and provides the "scaled value for the second group sub-band data" (col.

5, ln 7-8) which is then added to the first group subband data.

In reference to claims 11 and 31, ISO/IEC 11173-3 teaches the use of a polyphase filter bank for reconstruction of PCM data from subband data but does not teach how a phase positioning filter might modify subband data so that it is also inclusive of positional queues.

Claims 12, 20, 21 and 32 teach transmitting a null output template if the next frame of mixed subband data and scale factors is not ready to provide for seamless generation of output frames and maintain synchronization with a multi-channel decoder. If the decoder does not receive SYNC words (included in the header) at regular intervals it will assume that the data is in PCM format and can be passed directly to the DACs. If this happens the compressed audio will appear as noise at the speakers until synchronization can be reacquired. Paneth sends a NULL KNOWLEDGE message when the transmitted message is not ready to keep the modem clear of erroneous data. (col. 49, line 19). Paneth states "All slots contain eight symbols of "null" transmission, the filter startup field, which enables the modem to purge it's receive filters in order to prepare for the new slot." (col. 17, ln 33-35). The transmission of null symbols to keep the modem clear is quite different from transmitting a null template with an inaudible signal to maintain sync with a decoder.

In reference to claim 16, looping is a standard gaming technique in which the same sound bits are looped indefinitely to create a desired audio effect. For example, a small number of frames of a helicopter sound can be stored and looped to produce a helicopter for as long as the game requires. In the time domain, no audible clicking

or distortion will be heard during the transition zone between the ending and the starting positions of the sound if the amplitude of the beginning and ends are complementary. This same technique does not work in the compressed audio domain. To solve this problem Applicant created audio components "having commencing input frames and closing input frames whose subband data has been preprocessed to ensure seamless concatenation with the commencing frame" for looped data. (p. 13, line 5 to p. 14, line 3). Everett at col. 6, ln 7-9 teaches "sliders 121-126 which control the delay length of a reverberation loop mixing the output of a channel into the input after a delay period." Everett's technique for reverberation loop mixing to create "echoes" is very different than Applicant's technique for providing looped data in the compressed domain.

In reference to claim 17, Applicant's bitstream is coded with fixed length codes (FLCs). The claim has been amended to specify that said header and bit allocation table being fixed from component-to-component, channel-to-channel and frame-to-frame "so that only the content of the scale factors and subband data are allowed to vary". The Examiner cites Buffer 22 of figure 2 in Everett as evidence of his use of FLCs. However, the use of a buffer is irrelevant to whether data is coded using fixed or variable length codes as evidence by Everett's teaching that "The final section SBD carries the sub-band data as *variable length fixed point values*". (col. 4, ln 16-17). Clearly, Everett's specific teaching of variable length codes teaches away from Applicant's use of FLCs.

In claim 33 as dependent from claim 1, the API generates the list in response to an action input received

by the HID and tracks the positional coordinates of the audio components on the list. (p. 7, ln 3-4, 24-30). For example, in a gaming application the user will perform any number of actions that cause the API to select different audio components and put them on the list. The API tracks their positional coordinates so that the audio components are properly mixed to position the audio in the 3D environment during playback.

For the reasons provided to traverse the rejections to independent claims 1, 17, 21, 27 and 29 and for the additional features they recite, the rejections of the dependent claims are traversed. Applicant reserves the right to further address the specific rejections raised by the Examiner if necessary and at the appropriate time to establish their patentability.

Conclusion

It is respectfully urged that the subject application is patentable over the cited references and is now in condition for allowance.

The Examiner is invited to call the undersigned at the below listed telephone number if, in the opinion of the Examiner, such a telephone conference would expedite or aid the prosecution and examination of this application.

Respectfully submitted,

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